Introduction

What is DMT and where is it used?

DMT is a form of Orthogonal Frequency Division Multiplexing. Effectively, information is coded and modulated by several different sub carriers. This is the same modulation scheme used in common DSL channels. Specifically, at frequencies above those reserved for speech, there exist several hundred channels of the equal bandwidth used to transmit data to and from the internet. This is very convenient for phone users, as these DMT channels can be easily filtered out using a simple low-pass filter, preventing any possible interference from connecting to the internet while talking on the phone.

Why transmit over an acoustic channel?

The acoustic channel, and the audible frequencies contained therein, was chosen as a test area for our DMT project simply because of its accessibility. No complex or expensive hardware is necessary to transmit over it, just a standard computer speaker and microphone setup.

What can DMT do for us?

As will be revealed in later modules, DMT modulation offers many features that protect our signal's information from being distorted by the channel. Specifically, by transmitting our information over several subcarriers at the same time instead of just one carrier, we can use the frequency response of the system to see which carriers are being attenuated and likewise increase the gain on those channels or get rid of them altogether. You will discover, as we did, how difficult that process turned out to be.

The Problem

Creating a DSL Modem

Our goal is to use Discrete Multi-Tone modulation to transmit a text message over the audible range of frequencies in an acoustic channel. We are creating a DSL modem that transmits through the air.

To do this, we will observe the frequency characteristics of the channel, and use that information to equalize our received transmission in hope to preserve the maximum amount of content from the signal. As in any engineering problem, we are constantly striving to push the data-rate of our system, while minimizing the occurrence of errors. Here we go!

The Transmitter

The Channel

Receiver

Results and Conclusions

Transmitter

Text to Binary Conversion

The first step is to convert our information into binary. We used the sentence “hello, this is our test message,” repeated four times, as our text message. To get it into binary, we used standard ASCII text mapping.

hello = 01101000 01100101 01101100 01101100 01101111

Series to Parallel

The next step is converting this vector of zeros and ones into a matrix. The vector is simply broken up into blocks of length L, and each block is used to form column of the matrix.

Constellation Mapping

Now the fun begins. The primary method of modulation in DMT is by inverse Fourier Transform. Although it may seem counterintuitive to do so, by taking the inverse Fourier Transform of a vector or a matrix of vectors, it effectively treats each value as the Fourier coefficient of a sinusoid. Then, one could transmit this sum of sinusoids to a receiver that would in turn take the Fourier Transform (the inverse transform of the inverse transform, of course) and retrieve the original vectors.

But instead of taking the transform of our vectors of zeros and ones, we first convert bit streams of length B to specific complex numbers. We draw these complex numbers from a constellation map (a table of values spread out along the complex plane). See the figure below for an example of a 4 bit mapping.

Constellation Mapping Table

Bit Sequence	Constellation Value	Bit Sequence	Constellation Value
0000	.354 + .354j	1000	1
0001	.707	1001	.707 + .707j
0010	.707j	1010	j
0011	-.354 + .354j	1011	-.707 + .707j
0100	-.707j	1100	-1
0101	.354 - .354j	1101	-.707 - .707j
0110	-.354 - .354j	1110	-j
0111	-.707	1111	.707 - .707j

This table shows which bit stream is mapped to which complex value.

Signal Mirroring and Inverse Fourier Transform

Why would we do that, you might ask. Doesn't converting binary numbers to complex ones just make things more complicated? Well, DMT utilizes the inverse Fourier Transform in order to attain its modulation. So taking the IFFT of a vector of complex numbers will result in a sum of sinusoids, which are great signals to be sending over any channel (they are the eigenfunctions of linear, time-invariant systems).

But before taking the inverse transform, the vectors/columns of the matrix must be mirrored and complex conjugated. The Inverse Fourier Transform of a conjugate symmetric signal results in a real signal. And since we can only transmit real signals in the real world, this is what we want.

Cyclic Prefix

If we were transmitting over an ideal wire system, we would be done at this point. We could simply send it over the line and start demodulating. But with most channels, especially our acoustic one, this is not the case. The channel's impulse response has non-zero duration, and will therefore cause inter-symbol interference in our output.

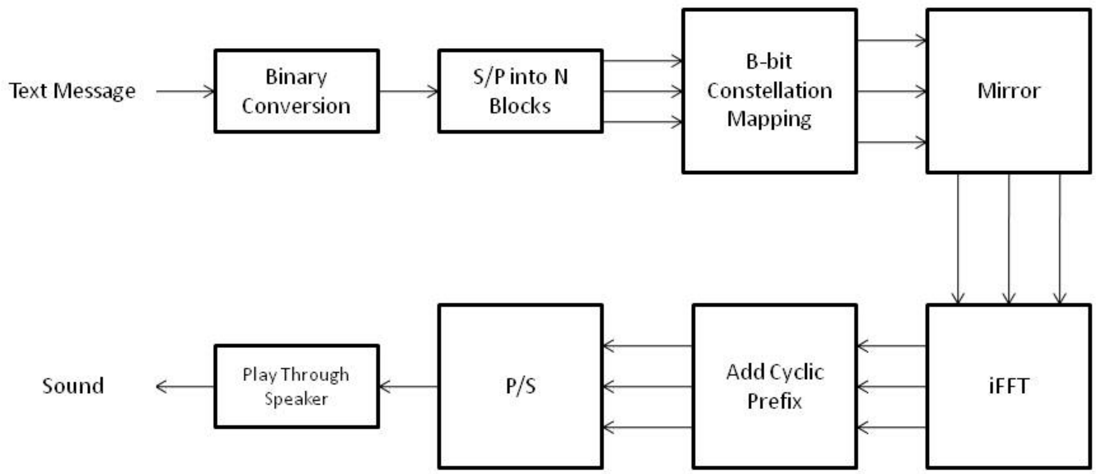
Intersymbol interference occurs during the convolution of the input and impulse response. Since the impulse response has more than a single value length, it will thus cause one block's information to bleed into the next one.

To prevent this, we added what is called a cyclic prefix to each block. As long as the length of the cyclic prefix is at least as long as the impulse response, it should prevent ISI. However, it has a secondary effect as well. We created the prefix by adding the last N values of each block (where N is the length of the response) to the beginning, preserving the order. Doing this effectively converts the linear convolution of the impulse response with the block sequence to circular convolution with each block separately, since there will now be the "wrap-around" effect. This will be handy later when we start characterizing the channel, since circular convolution in time is equivalent to multiplication of DFT's in frequency.

0001011001101000 1 => 0100010001011001101000 1

The first six bits in the second bit stream, 010001, is the cylcic prefix. Note that although these values are binary, they could essentially range from -1 to 1 since they sample the sinusoid sum that was formed after inverse Fourier Transforming.

Please see the block diagram below. It summarizes the entire transmission process covered above.
Transmission Block Diagram



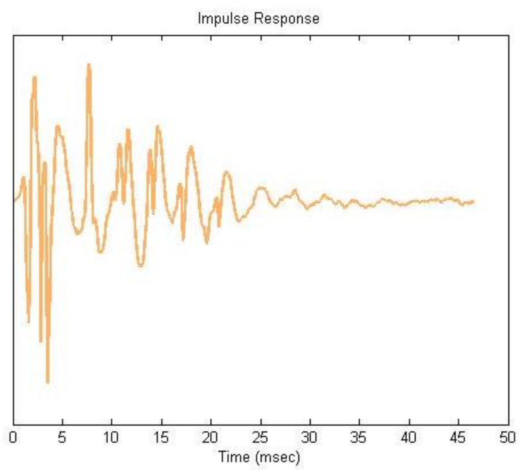
This diagram shows the all of the components and flow of our transmission system.

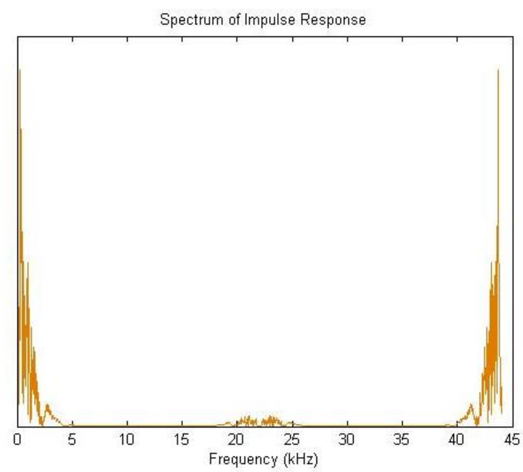
The Channel
After transmission our signal enters the acoustic channel, join us as we witness what unfolds

The Channel

To characterize the channel, we input an impulse by recording the tapping of the mic with our fingers. We then played that sound through the speaker and recorded the response with the mic. The signal is below, along with its spectrum.

Impulse Response of the Channel and its Spectrum



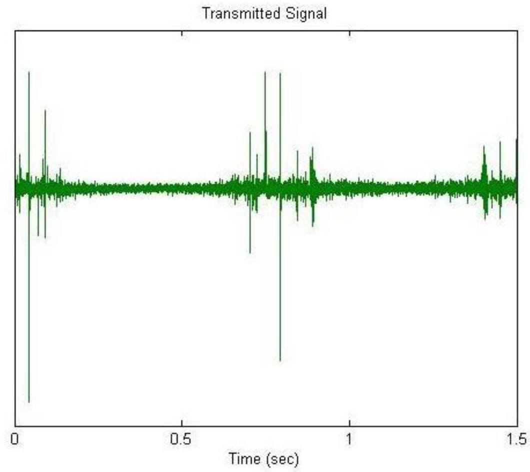


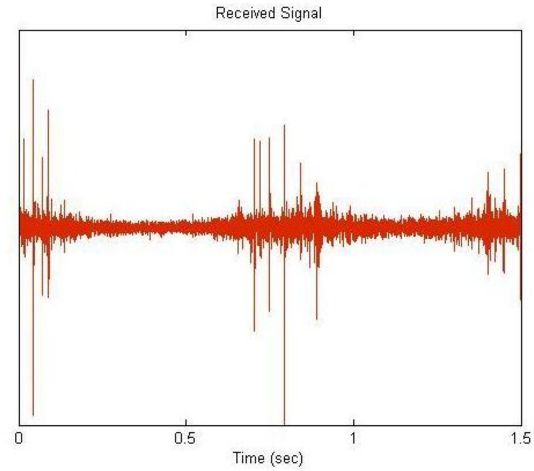
These graphs characterize the channel that we are transmitting through

We did this in preparation for the receiving end of the system to divide the received signal's FFT by the impulse response's FFT.

Below are plots of our transmitted and received signals, along with their spectrums. You will notice a great similarity between the signals in time, however a distinct difference in frequency. Unfortunately, this loss in frequency will translate to a loss of information.

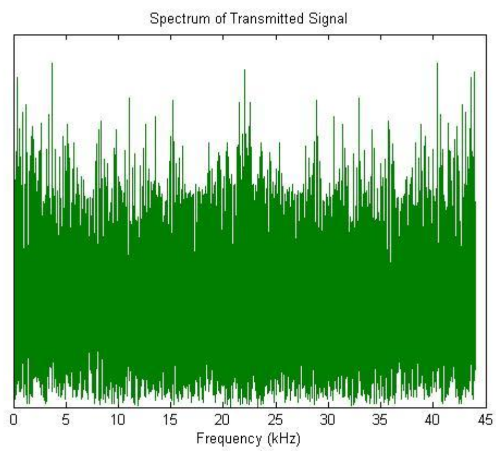
Transmitted and Received Signals in the time domain

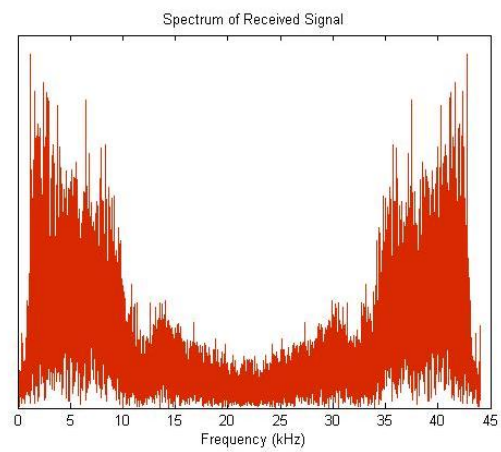




These are the signals in time that we transmitted (green) and that we received (red). As you can see they look very similar, and take it from us, they also sound similar.

Transmitted and Received Signal Spectrums





The green spectrum is of the signal we transmitted, and the red is the spectrum of the signal we received. We see a much bigger visual difference than we did in the time domain.

Above are plots for our transmitted and received signals. Here we used a block length of half the duration of the signal and sent it through the air at

44.1 kHz.

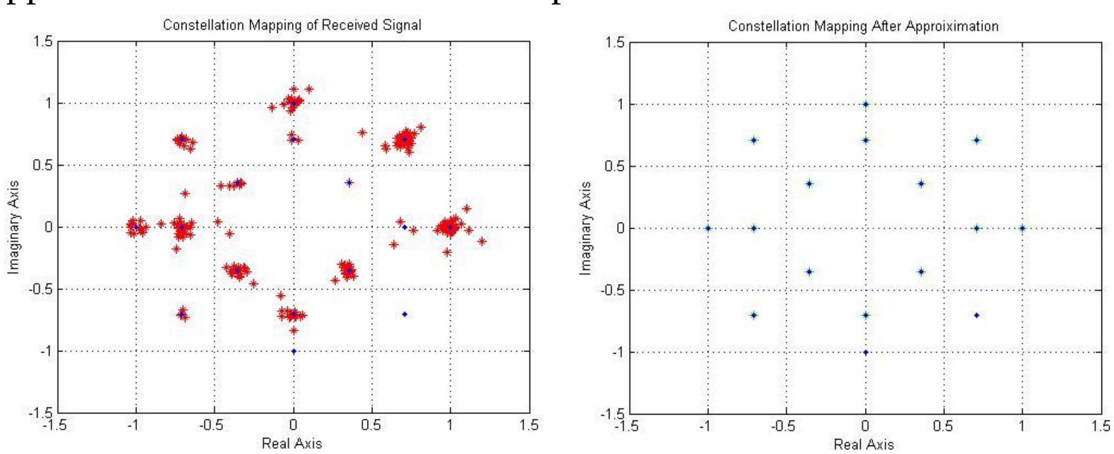
Receiver

Decoding the Transmission

Since the receiver has full knowledge of all the steps taken to transmit, the reception process is the exact inverse of transmission. The only difference is the addition of the channel equalization described in the previous part. To get back the information we originally sent, we simply:

- Take the FFT of the reception and divide it by the FFT of the impulse response. Then iFFT it back.
- Remove the cyclic prefix
- Take the Fourier Transform
- Demirror the vector
- Approximate each received value to nearest point in constellation and map them back to the original bit sequences. See figure below for example in 4 bit approximation.
- Convert the binary series back to ascii letter equivalents.

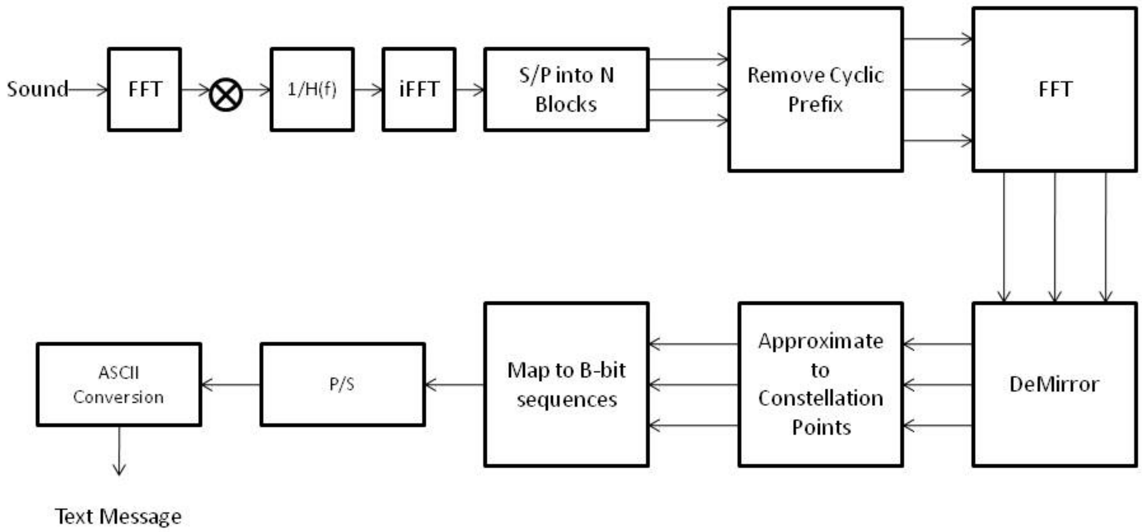
Approximation of Constellation Map



The map on the left was approximated to the one on the right with a 2.15% percent error.

Please see the block diagram below. It summarizes the reception process.

Receive Block Diagram



This diagram shows the all of the components and flow of our receiver system.

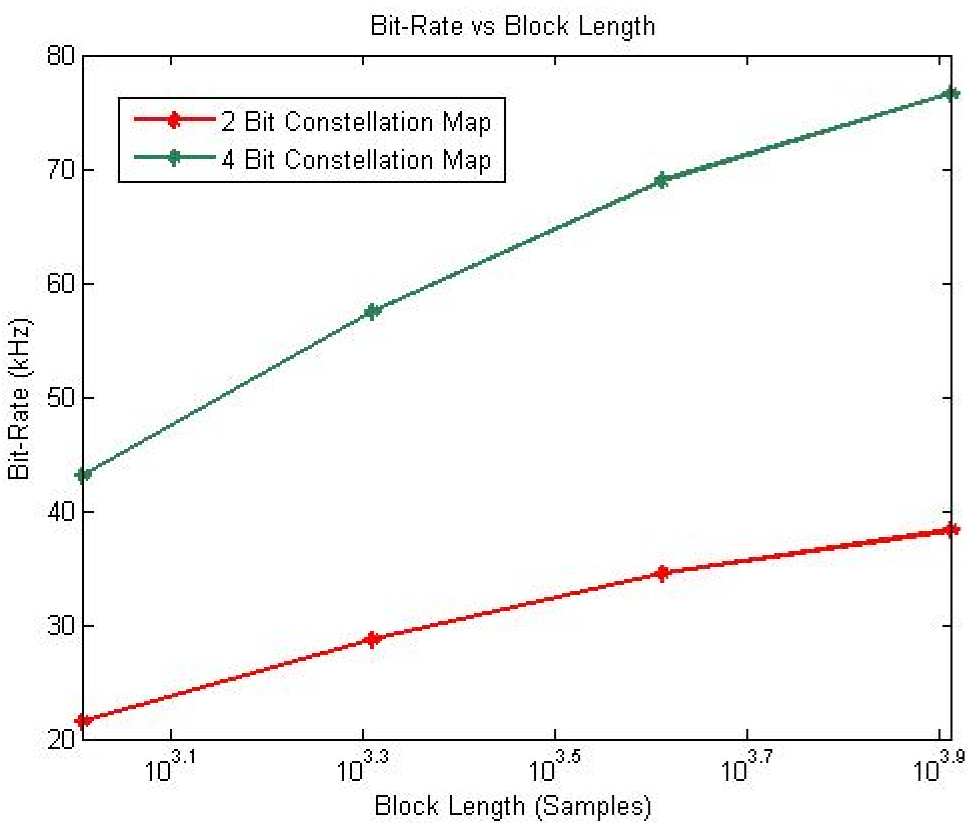
Results and Conclusions
The outcome of our DMT transmission system

Results

Unfortunately, our microphone-speaker system was not successful in transmitting a text message. The measured transfer function seemed reasonable since it modeled a low-pass filter. But it was ineffective in equalizing our received signal. This is most likely because the channel added far too much noise, in addition to attenuating many of the frequencies beyond recovery.

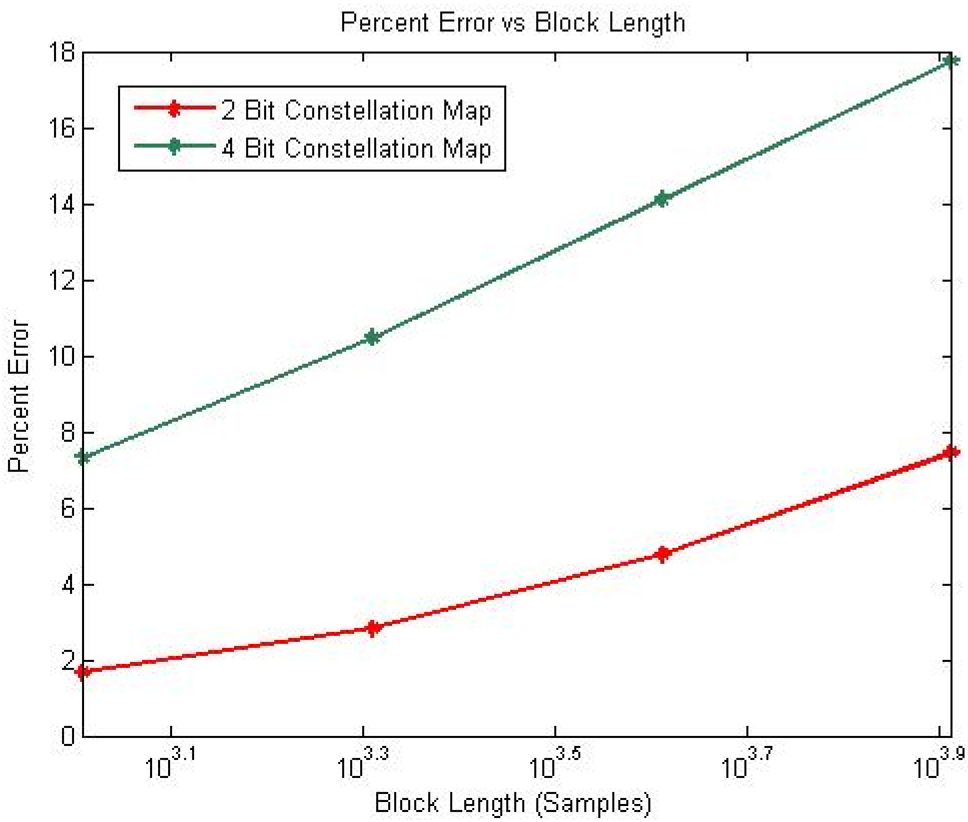
Since we were unable to acquire the desired results on bit-rate maximization and error minimization in the acoustic channel, we created an artificial channel, using our observed frequency response plus Gaussian noise. Modeling this channel in Matlab produced notable results. See the figures below.

Bit-rate vs Block Length (2 and 4 bit)



This graph illustrates the fact that data rate increases as we increase block length. It also shows that a 4-bit scheme has twice the data rate of a 2-bit scheme, as one would expect.

Percent Error vs Block Length (2 and 4 bit)



This graph illustrates the fact that as block length increases so will the amount of errors. Also we see that the 4-bit scheme has a much greater amount of errors than the 2-bit.

These figures indicate that both bit-rate and error-rate go up as block length increases. This makes sense since increasing the block length increases the number of channels (Taking the iFFT of a longer signal produces more unique sinusoids.). Squeezing more sinusoid carriers over the same bandlimited channel (0-22kHz) should result in more errors in demodulation, while transmitting more bits at the same time. It also makes

sense that the 4 bit constellation mapping yielded higher bit-rates and error percentages since each sinusoid carries more information, yet can more easily be approximated to the wrong constellation point.

The next project dealing with Discrete Multi-Tone modulation in the acoustic channel should certainly involve a more professional recording system.

Our Gang
What what!

Gang Members

Dangerous Brian Viel –Electrical and Computational Engineering

Soarin’ Dylan Rumph – Electrical and Computational Engineering

Acknowledgements

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